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PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
(Attorney Docket No. 12441ROUS02U)

In the Application of: Periyalwar et al.

Serial No.: 09/834,104

Filed: April 12, 2001

For: Frame Structures Supporting Voice or Streaming Communications with High Speed Data Communications in Wireless Access networks

§ Group Art Unit: 2661
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§ Examiner: Unknown
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Bruce E. Garlick

SUBMITTAL OF PRIORITY DOCUMENT (FOREIGN PATENT APPLICATION)

Assistant Commissioner for Patents
Washington, D.C. 20231

Dear Sir:

Enclosed herewith is a Certified Copy of Canadian Patent Application 2,305,082, filed on April 12, 2000, to which the present application claims priority.

Respectfully submitted,

Date: July 18, 2001

By: _____
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This is to certify that the documents attached hereto and identified below are true copies of the documents on file in the Patent Office.

Specification and Drawings, as originally filed, with Application for Patent Serial No: **2,305,082**, on April 12, 2000, by **NORTEL NETWORKS CORPORATION,** assignee of Shalini S. Periyalwar, Leo L. Strawczynksi and Wen Tong, for "Method for Efficient Support of Simultaneous Voice and Data in Adaptive High Data Rate Wireless Access Systems".

S. Gregoire
Agent certificateur/Certifying Officer

May 30, 2001
Date

Canada

OPIC CIPO

METHOD FOR EFFICIENT SUPPORT OF SIMULTANEOUS VOICE AND DATA IN ADAPTIVE HIGH DATA RATE WIRELESS ACCESS SYSTEMS

TECHNICAL FIELD

The present invention relates to a high data rate cellular systems and in particular to a method for efficient support of simultaneous voice and data in adaptive high data rate wireless access systems.

BACKGROUND ART

High data rate (HDR) cellular systems designed primarily for date users have been proposed by Qualcomm (corporate name). Detailed specifications of the HDR cellular systems are disclosed in the submission "HDR Air Interface (HAI) Specification" of 2000.03.20. There is, however, no convenient method of providing simultaneous access to voice calls within the same system. Any voice calls are deflected to the peer cdma2000/IS-95 system. Also, the system requires that both HDR and cdma2000 be deployed for simultaneous support of voice and date calls.

A possible solution for the proposed HDR cellular systems is to deflect all voice calls to the peer IS-95/cdma2000 network. While this is proposed to maximize the efficiency of the HDR cellular system, it is not practical, particularly when there are users requiring simultaneous connections to voice and data calls.

SUMMARY OF THE INVENTION

It is an object of the present invention to provide a method which provides data call customers with the ability to concurrently carry on a voice call whiteout directing the voice call over a complementary or peer network. As a further advantage, the voice call is carried by the same high speed access network as the data call without adversely affecting the efficiency and speed of the data traffic.

According to the most general aspect of the present invention, there is provided a method for providing data call customers with the ability to concurrently carry on a voice call whiteout directing the voice call over a complementary or peer network, the method comprising steps of: identifying voice packets by a unique preamble identifier to enable voice user to easily identify voce packet and the data rate at which it is transmitted; stopping processing voice packets

once their voice frame is received, efficient scheduler; minimizing the number of slots used to transmit voice; scheduling voice packets at the first opportunity on start of a new (e.g., 20 ms) frame;
multiplexing users with similar channel and interference conditions (C/I).

The invention presumes that low-bit rate voice is encrypted at the link level.

It is to provide a unified solution for transmission of both low bit rate real time as well as delay tolerant non-real time signals within the same system.

BRIEF DESCRIPTION OF THE DRAWINGS

An embodiment of the present invention will now be described by way of example only with reference to the accompanying drawings in which:

Figures 1 and 2 show examples of voice packet encoder format.

DETEAILED DESCRIPTION

The present invention is directed to be applied into the proposed HDR cellular systems described in "HDR Air Interface (HAI) Specification" of 2000.03.20. The specifications are incorporated herein by reference. There is, however, no convenient method of providing simultaneous access to voice calls within the same system.

The present invention is as follows:

Support of Simultaneous Voice and Data

Reverse Link:

- Voice as per 1xRTT

Forward link:

- Voice support for simultaneous high data rate users only
  - Support limited number of voice subscribers
  - Other voice/low data rate users routed to 1xRTT carriers
- Efficient traffic management through segregation of voice and data traffic

- multiplexing allows efficient bandwidth management to accommodate multiple voice users

- Objective - to provide voice coverage comparable to IS-95/1xRTT
  - Qualcomm analysis concludes HDR coverage extends well beyond voice (14.4 kbps voice and a 614.4 kbps data rate have similar range for a single receive antenna)
  - 38.4 kbps data @ 100% power allocation is roughly equivalent to 9.6 kbps voice @ >25% power allocation
  - Conclude - Minimum data rate required to support voice services > or = 76.8 kbps
- Define Traffic Channel Encoder Packet Format for Voice

Tables 1 and 2 show forward channel modulation parameters for voice.

Voice packet preamble structure:

- Preamble is time multiplexed into the traffic channel stream at the beginning of the first slot of the voice packet being transmitted
- Voice packet is identified by a unique 32-ary Walsh function, W^{32}_{voice}, repeated several times depending on the data rate of the packet and BPSK modulated on the in-phase modulation phase (I)
- Explicit Data Rate Indication (EDRI) is represented by a 4 bit EDRI symbol. Each EDRI symbol maps into a 16-ary Walsh orthogonal code word repeated 2x as often as the voice packet identifier and BPSK modulated on the quadrature modulation phase (Q)

Note:

- Assuming the power on I and Q are equal then the above preamble structure has a 3 dB penalty relative to the HAI specification. However, the specification appears to include a significant protection margin.

Alternate voice packet preamble structure:

- Preamble is time multiplexed into the traffic channel stream at the beginning of the first slot of the voice packet being transmitted
- Voice packet and data rate of the packet are identified by a 32-ary Walsh function, $W^{32}_{voice/rate}$, repeated several times depending on the data rate of the packet and BPSK modulated on the in-phase modulation phase (I) If the sign of the Walsh function is also used then 4(5) Walsh codes are required to identify voice packets with up to 8(10) data rates.

Voice Packet Encoder Format

- Multiplex several voice calls per Forward Voice Packet (similar to AAL2)
-
  - Pointer Field (optional)
    - indicates the bit position (relative to the end of the pointer) of the first complete voice frame in the packet (Pointer is not required for the first voice packet in a 20 ms Frame)
  - Rate/length identifier
    - 9.6 kbps, R=1 (172 bits), 1/2 (80 bits), 1/4 (40 bits), 1/8 (16 bits)
    - 14.4 kbps, R= 1 (267 bits), 1/2 (125 bits), 1/4 (55 bits), 1/8 (21)

Scheduler:

- Objective - minimize the number of slots per 20 ms required to transmit voice
- Voice packets are transmitted at the first opportunity after a 20 ms frame pulse until transmission of the voice is complete or the maximum number of slots assigned to voice are transmitted
- Sets the voice packet data rate
- Multiplexes multiple voice users onto a voice packet
  - Assigns voice users to voice packets such that the data rate requested by the user is > or = voice packet data rate

Voice Packet Encoder Format

See Figures 1 and 2.

Forward Channel Modulation Parameters for Voice

Data Rates (kbps)

Parameter	76.8	102.4	153.6	204.8	307.2
Bits per Encoder Packet	384	384	384	768	384
Slots per Encoder Packet	4	3	2	3	1
Encoder Packet Duration (ms) chips	5.0 6144	3.75 4608	2.5 3072	3.75 4608	1.25 1536
Preamble Puncture Duration (chips)	512	384	256	192	128
Code Rate	1/4	1/4	1/4	1/4	1/4
Modulation Type	QPSK	QPSK	QPSK	QPSK	QPSK

TABLE 1

Forward Channel Modulation Parameters for Voice

Data Rates (kbps)

Parameter	614.4	921.6	1228.8	1843.2	2457.6
Bits per Encoder Packet	768	1152	1536	2304	3072
Slots per Encoder Packet	1	1	1	1	1
Encoder Packet Duration ms chips	1.25 1536	1.25 1536	1.25 1536	1.25 1536	1.25 1536
Preamble Puncture Duration (chips)	64	64	64	64	64
Code Rate	1/4	1/4	1/4	1/4	1/4
Modulation Type	QPSK	QPSK	QPSK	8 PSK	16 PSK (QAM)

TABLE 2

The embodiments of the invention in which an exclusive property or privilege is claimed are defined as follows:

1. A method for providing data call customers with the ability to concurrently carry on a voice call whiteout directing the voice call over a complementary or peer network, the method comprising steps of:

identifying voice packets by a unique preamble identifier to enable voice user to easily identify voce packet and the data rate at which it is transmitted;
stopping processing voice packets once their voice frame is received, efficient scheduler;
minimizing the number of slots used to transmit voice;
scheduling voice packets at the first opportunity on start of a new (e.g., 20 ms) frame;
multiplexing users with similar channel and interference conditions (C/I).

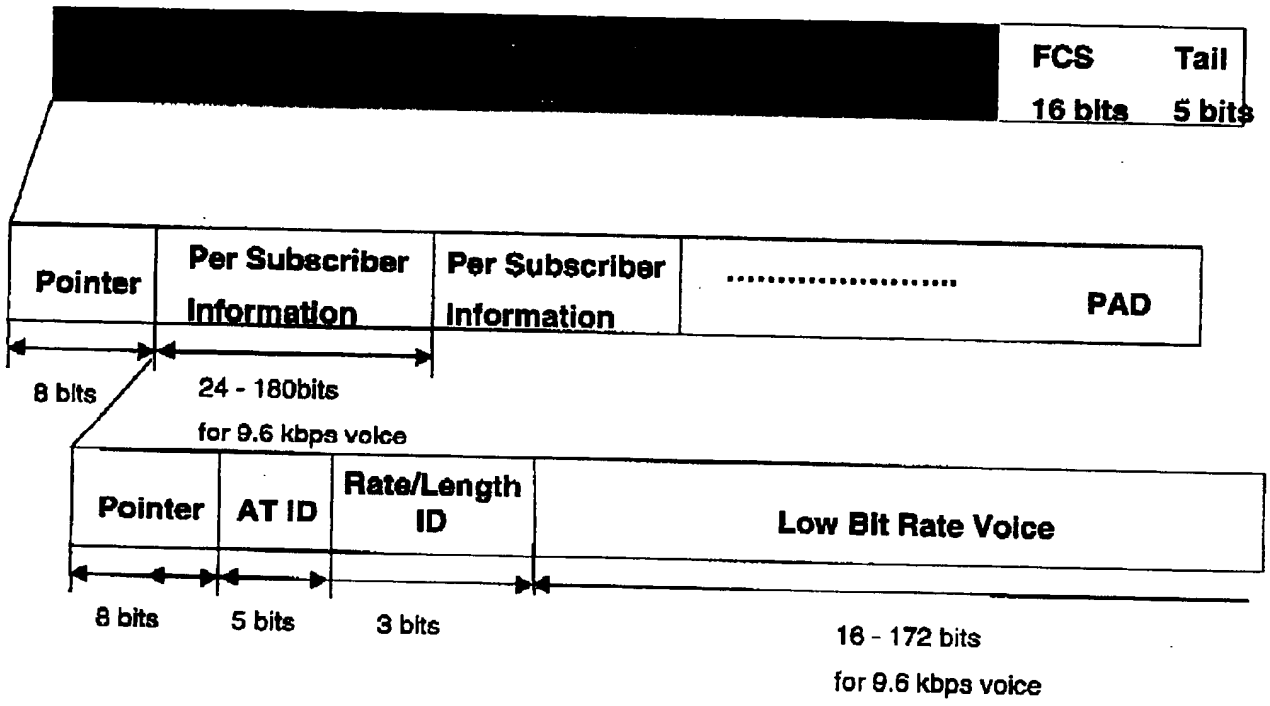
FCS
16 bits
Tail
5 bits
Pointer
Per Subscriber Information
Per Subscriber Information
.....
PAD
8 bits
24 - 180bits
for 9.6 kbps voice
Pointer
AT ID
Rate/Length ID
Low Bit Rate Voice
8 bits
5 bits
3 bits
16 - 172 bits
for 9.6 kbps voice

FIG. 1

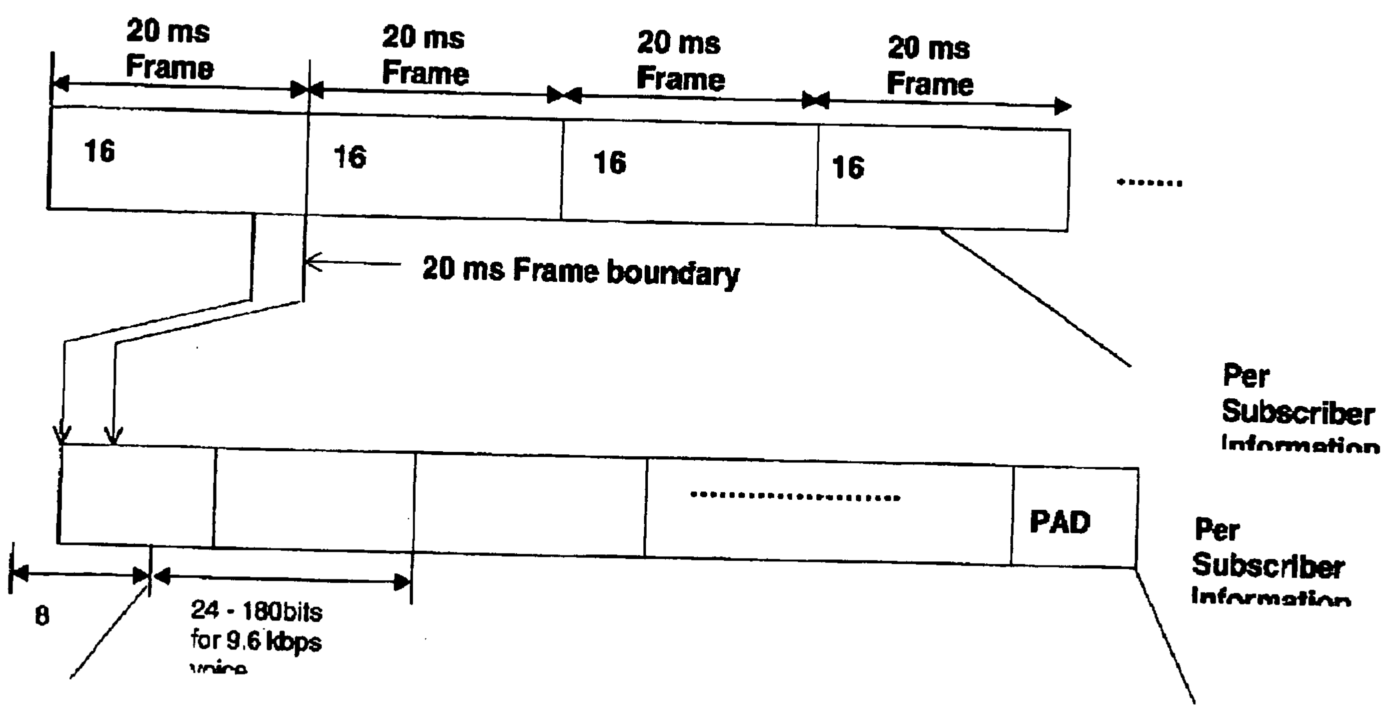

FIG. 2